

1           **METHOD OF IMPROVING AUDIO PERFORMANCE AND**  
2   **POWER UTILIZATION OF A PORTABLE AUDIO DEVICE WITH**  
3   **ELECTRONIC ANTI-SHOCK SYSTEM (EASS)**

4   **BACKGROUND OF THE INVENTION**

5   1. Field of the Invention

6           The present invention is related to a method of improving performance  
7   and power utilization of a portable audio device fitted with an electronic anti-  
8   shock system (EASS), and more particularly to a method of improving  
9   performance and power utilization of a portable audio device such as CD player  
10   by employing MPEG, a high compression rate algorithm, to encode/decode  
11   audio signals so as to increase the capacity of audio data stored in a temporary  
12   memory, as a result extending the buffering time, and realizing power saving.

13   2. Description of Related Art

14           CD players usually have an electronic anti-shock system (EASS) or  
15   equivalent buffering device that creates a data buffer in the signal processing  
16   path between the data retrieval lens and the audio signal processor to prevent  
17   interruptions in audio playback. If CD players are subjected to shocks or  
18   vibrations during reading of audio data, audio signal processing will be  
19   interrupted, and the quality of audio output will be degraded accordingly.

20           A conventional EASS has the function of a CODEC, a coding-decoding  
21   device that converts audio signals into digital bit streams and back again, and the  
22   basic structure is shown in Fig. 3, in which separate data paths are used to  
23   process audio data going through the right and left audio channels. The input  
24   pulse code modulation (PCM) signals going through the right and left channels

1 are respectively processed by a pair of ADPCM encoders (71) (71'), and then  
2 saved in a pair of memory devices (DRAM) (72) (72'). For faithful reproduction  
3 of the sound, audio compressed data is read from the above memory devices  
4 (DRAM) (72) (72'), and then passed to a pair of corresponding ADPCM  
5 decoders (73) (73') to restore to the original PCM signals for the right and left  
6 channels. Thereafter, the decoded audio signals in the right and left channels are  
7 simultaneously passed to an audio signal processor (74) for output through a  
8 speaker. Basing on the above structure, the above mentioned temporary memory  
9 in the audio processing path has a buffering effect that can somewhat prevent  
10 vibration-caused interruptions during audio playback.

11           However, the conventional electronic anti-shock system (EASS)  
12 employs the adaptive differential pulse code modulation (ADPCM) for encoding  
13 and decoding, by which the waveform of the analog signals are sampled at a  
14 fixed frequency.

15           Generally, audio data have to be compressed before they can be saved in  
16 a memory. The relationship between data size and the compression rate will be  
17 explained hereunder, as it concerns the utilization of memory resources. For  
18 example, the waveform of 12 bit/sample is compressed with a 3:1 ratio to  
19 become 4 bit/sample, and when reading out data from memory, the data is  
20 decompressed with 1:3 ratio to restore to the original audio format of 12  
21 bit/sample for sound reproduction. Therefore, the memory used to save the audio  
22 data is less than that with standard PCM codes, but the compression rate is not  
23 adequate for saving large amounts of audio data in the limited memory of a CD  
24 player.

1           The compression coding scheme for the above-mentioned conventional  
2 ADPCM encoder (71) can be implemented in either 3-bit mode or 4-bit mode.  
3 Since all current audio equipment has at least two audio channels, the 4-bit  
4 operation mode is selected in this example for calculation of the bit rate with a  
5 sampling frequency of 44.1 KHz:

6            $4(\text{bits}) \times 44100 \times 2 (\text{number of audio channels}) = 352,800 \text{ Kbps}$

7           If the operation is in 3-bit operation mode, the bit rate is:

8            $3(\text{bits}) \times 44100 \times 2 (\text{number of audio channels}) = 264,600 \text{ Kbps}$

9           If the memory installed in the above anti-shock system is 16M bits, as in  
10 the present example, the required buffering time for the two operation modes can  
11 be:

12           4-bit mode:  $16,000,000 \div 352,800 = 45.35 \text{ (sec)}$

13           3-bit mode:  $16,000,000 \div 264,600 = 60.46 \text{ (sec)}$

14           From the above explanation, the buffering time of the EASS is  
15 dependent on the bit rate and the memory capacity. The data saving operation in  
16 the EASS has to be sustained for a longer duration if the performance of a CD  
17 player having EASS is to show any noticeable improvement.

18           Within the constraint not to increase a DRAM memory, the only way to  
19 improve the performance of CD player is to increase the bit rate in the audio  
20 compression. If the compression rate is increased, the performance of EASS can  
21 be improved without using additional memory.

22           However, the ADPCM algorithm primarily is not primarily designed for  
23 audio data compression, as the above compression rate cannot support high  
24 capacity storage in limited memory storage of a portable CD player. Satisfactory

1 buffering will need large amount of DRAM memory, which is quite difficult for  
2 a compact sized CD player, not to mention the increased costs.

3 From the foregoing, it is quite clear that ADPCM cannot satisfy the  
4 present requirement of data conversion with high compression, but there are  
5 some more advanced compression algorithms, such as the MPEG layer I and II,  
6 which are able to produce reasonably acceptable sound quality with much better  
7 utilization of memory than ADPCM. Audio data with double or triple larger size  
8 can be saved in the same amount of memory space as compared with the  
9 conventional ADPCM-based systems.

10 For portable CD players, another benefit of using MPEG in the EASS is  
11 the power saving feature. When the audio data is read from the data buffer, the  
12 servomotor of the CD player is kept in a suspended mode. Therefore, the longer  
13 the CD servo can be suspended by the operation in the data buffer, the less the  
14 system power is used.

15 Therefore, the present invention attempts to incorporate an audio  
16 compression algorithm having high compression rate in the EASS to attain the  
17 most desirable balance point between audio performance, power management,  
18 and costs.

## 19 SUMMARY OF THE INVENTION

20 The main object of the present invention is to provide a method for  
21 improving the audio performance of a portable CD player by adopting the  
22 Moving Picture Experts Group (MPEG), a high compression rate algorithm, in  
23 the electronic anti-shock system (EASS), so as to increase the storage capacity of  
24 the temporary memory and extend the buffering time.

1           When PCM signals are received, the EASS use an MPEG compression  
2 algorithm with high compression rate to convert the PCM signals to digital  
3 values in the form of data streams and save them in the temporary memory.  
4 Conversely, when the audio compressed data are read out from the temporary  
5 memory, EASS uses the above compression algorithm to convert the audio  
6 compressed data to restore to the original PCM format for sound reproduction.

7           The precondition for using the MPEG compression algorithm is that the  
8 sound reproduction of the CD player shall closely resemble the sound quality in  
9 the information medium (CD). Since the compression rate using MPEG is a  
10 multiple of the conventional ADPCM, a double or triple amount of audio data  
11 can be saved in the temporary memory, thus the buffering time of the EASS can  
12 be lengthened considerably. Therefore, the system can effectively prevent  
13 vibration-caused interruptions during audio playback.

14           The above audio compression format can be either MPEG 1 or MPEG 2.

15           The secondary object of the present invention is to provide an improved  
16 electronic anti-shock system (EASS) for portable CD players that is capable of  
17 realizing power saving. During the data operation in the buffer memory, the CD  
18 servomotor can be kept in the suspended mode using minimal power, as  
19 compared with the active mode, in which the servomotor becomes a major  
20 power user in the system.

21           Other objectives, advantages and novel features of the invention will  
22 become more apparent from the following detailed description when taken in  
23 conjunction with the accompanying drawings.

#### 24 BRIEF DESCRIPTION OF THE DRAWINGS

1           Fig. 1 is a comparative chart of the compression rates between MPEG  
2 layer II and layer III showing data sizes in different stages;

3           Fig. 2 is a system block diagram of the present invention; and

4           Fig. 3 is a system block diagram of a conventional electronic anti-shock  
5 system (EASS).

#### 6   DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

7           The disclosed electronic anti-shock system (EASS) is created by  
8 converting the input PCM signals to bit streams and saving them in a temporary  
9 memory to create a data buffer and after a predetermined amount of time the  
10 audio data are read out from memory and converted back to the original PCM  
11 format for sound reproduction.

12          According to the present invention, a high audio compression algorithm  
13 such as MPEG is used to convert the input PCM signals to audio compressed  
14 data and save them in temporary memory, and after a certain time the saved  
15 audio data are read out from the temporary memory and converted by the same  
16 audio compression algorithm back to the original PCM format;

17          Since the compression rate tends to be inversely related to the quality of  
18 sound reproduction, the adoption of a high compression rate has to be very  
19 carefully considered to not affect the reproduction quality during playback. The  
20 advanced compression algorithm, such as the Moving Picture Experts  
21 Group(MPEG)/layer II and III, is able to produce acceptable sound quality using  
22 much less memory than in an ADPCM, provided that the sound reproduction of  
23 the CD player shall closely resemble the sound quality in the recording medium  
24 (CD). The compression rates of MPEG Layer II, Layer III are shown in Fig. 1. It

1 is clear that the MPEG Layer III is even more powerful occupying even smaller  
2 memory.

3 For a CD player having EASS, the application of Layer II and Layer III  
4 will produce different results, because the buffering time for implementation  
5 with Layer III will be longer than that with Layer II given the same amount of  
6 memory.

7 For example, as shown in Fig. 1, using 160K bits compression rate with  
8 LAYER II, and 128K bits with LAYER III; and applying this on a DRAM  
9 memory with a capacity of 16M bits, the buffering time can thus be computed as  
10 follows:

11 LAYER II :  $16,000,000(\text{bits}) \div 160,000(\text{bps}) \doteq 100 \text{ (sec)}$

12 LAYER III:  $16,000,000(\text{bits}) \div 128,000(\text{bps}) \doteq 125 \text{ (sec)}$

13 Compared with the ADPCM algorithm in the 3-bit and 4 bit modes, the  
14 buffering time will be extended two to three times longer, thus the efficiency of  
15 the EASS can be improved considerably preventing vibration-caused  
16 interruptions in audio playback.

17 The structure of the present invention, as implemented in one preferred  
18 embodiment is shown in Fig. 2, comprises a MPEG encoder (10), a memory  
19 device (DRAM) (20), a DRAM controller (30), and an MPEG decoder (40).

20 The MPEG encoder (10) is used for converting PCM signals in the left  
21 channel (s<sub>l</sub>) and the right channel(s<sub>r</sub>) and applying the MPEG compression  
22 algorithm to produce audio compressed data streams.

23 The memory device (DRAM) (20) is used for temporarily keeping  
24 audio data en route to the audio signal processor, of which the input and the

1 output are respectively connected by a FIFO buffer (21) (22), and the input  
2 FIFO buffer (21) is connected to the output of the MPEG encoder (10).

3 The DRAM controller (30) is used for regulating the data flow to or  
4 from the memory device (DRAM) (20), wherein the DRAM controller (30) is  
5 respectively connected with the memory device (DRAM) (20) and two FIFO  
6 buffers (21) (22).

7 The MPEG decoder (40) is used for decoding the audio compressed data  
8 passed from the memory device (20), and restoring them to the original PCM  
9 format for sound reproduction, wherein the MPEG decoder (40) is connected to  
10 the memory device (DRAM) (20) through the FIFO buffer (22).

11 The above MPEG encoder (10) and MPEG decoder (40) may be in  
12 compliance with either MPEG 1 or MPEG 2 specifications.

13 The data processing operation under the above mentioned architecture is  
14 to be explained with reference to Fig. 2.

15 Input PCM signals of the left and right channel ( $s_l$ ) ( $s_r$ ) are passed to the  
16 MPEG encoder (10) to produce audio compressed data, wherein the data are  
17 temporarily saved in a static random access memory (SRAM) through a FIFO  
18 buffer (unnumbered), which enables a dynamic configuration module to conduct  
19 sideband coding, and then the audio data are further processed through  
20 quantizing and packetizing to produce a digital data stream representing the  
21 audio compressed data.

22 The data stream through the FIFO buffer (21) is written into the memory  
23 device (DRAM) (20) by means of the DRAM controller (30). Since the MPEG  
24 encoder (10) uses a high compression rate in the signal processing, the amount of



1 output data from the MPEG encoder (10) is considerably reduced, and the  
2 utilization of the memory device (DRAM) (20) can thus be improved.

3         Thereafter, the audio data, saved in the memory device (20) for a  
4 predetermined time, are read out and passed to the MPEG decoder (40) through  
5 the FIFO buffer (22), wherein the data are first depacketized to remove the  
6 encapsulation over the data, and then further through reverse quantizing and  
7 phase negation to restore to the original PCM format. After further  
8 reconfiguration and signal processing, the audio signals are played back over the  
9 speaker.

10         From the foregoing, it is apparent that the instrumentality of the present  
11 invention is to increase the compression rate of the EASS so as to increase the  
12 utilization of temporary memory. As a result, the system can prevent vibration or  
13 shock-caused interruptions during audio playback. However, the precondition to  
14 using the high compression rate algorithm is that the quality of sound  
15 reproduction of the CD player has to closely resemble the original recording  
16 level on the information medium (CD). If the above condition can be satisfied,  
17 the buffering time can be effectively lengthened, and the power saving can also  
18 be realized by increasing the time of the CD servomotor in the suspended mode.

19         When compared with the ADPCM compression algorithm, double or  
20 triple amounts of audio data can be saved in the same amount of memory, and  
21 the buffering time of the EASS can be lengthened considerably.

22         According to the present invention, the CD player having EASS is able  
23 to achieve the most desirable balance point between audio performance, power  
24 saving and low cost.

1           It is to be understood, however, that even though numerous  
2 characteristics and advantages of the present invention have been set forth in the  
3 foregoing description, together with details of the structure and function of the  
4 invention, the disclosure is illustrative only, and changes may be made in detail,  
5 especially in matters of shape, size, and arrangement of parts within the  
6 principles of the invention to the full extent indicated by the broad general  
7 meaning of the terms in which the appended claims are expressed.